(12) UK Patent Application (19) GB (11) 2 094 592 A
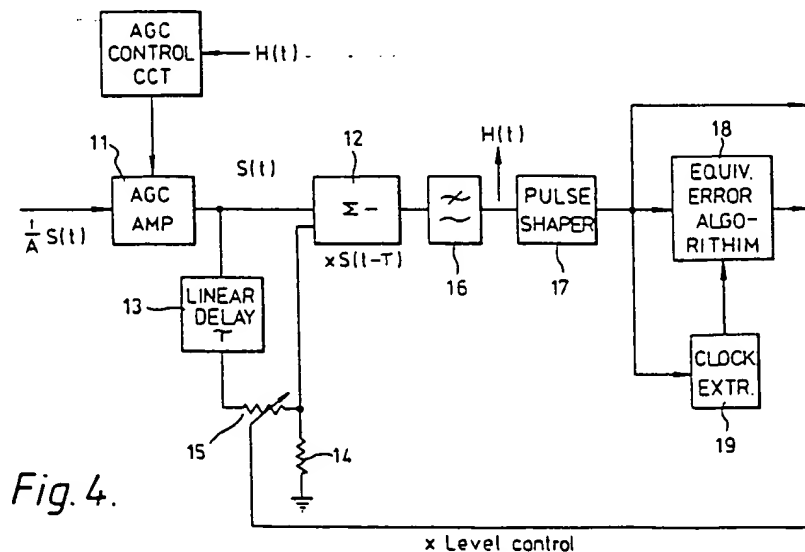
DOC

(54) Adaptive equaliser

(57) In a digital system, especially where PCM coded speech is conveyed over twisted pair cables optimised for speech, a pulse is received distorted, the distortion being especially bad on the trailing edge of the pulse. This decays slowly enough to cause inter-symbol interference with the next pulse.

This is overcome by applying the signal via an amplifier (11) direct to one input of a subtractor (12) and via a delay 13 and variable attenuator (14-15) to the other input of the subtractor. The delay is such as to put the delayed and attenuated version at the bit time for the next pulse: hence the interference due to the slow decay of the trailing edge is eliminated.
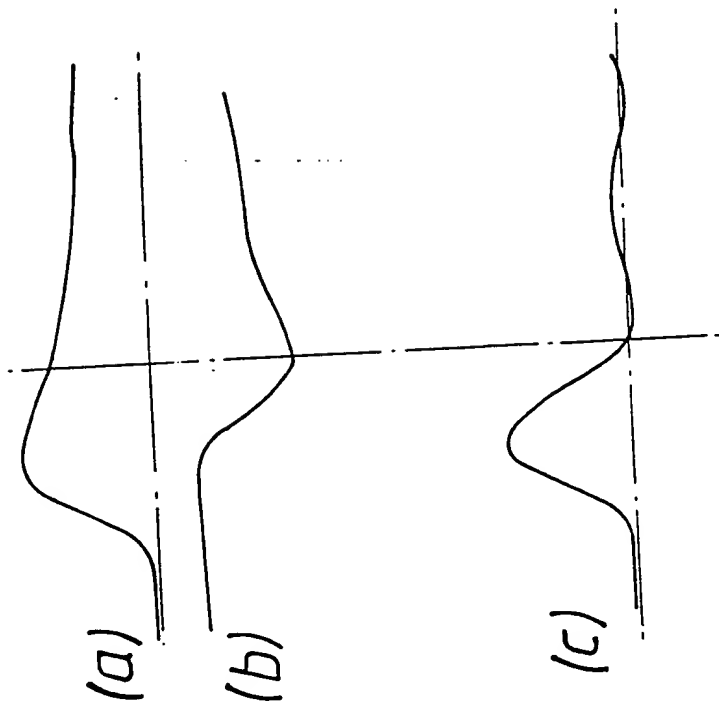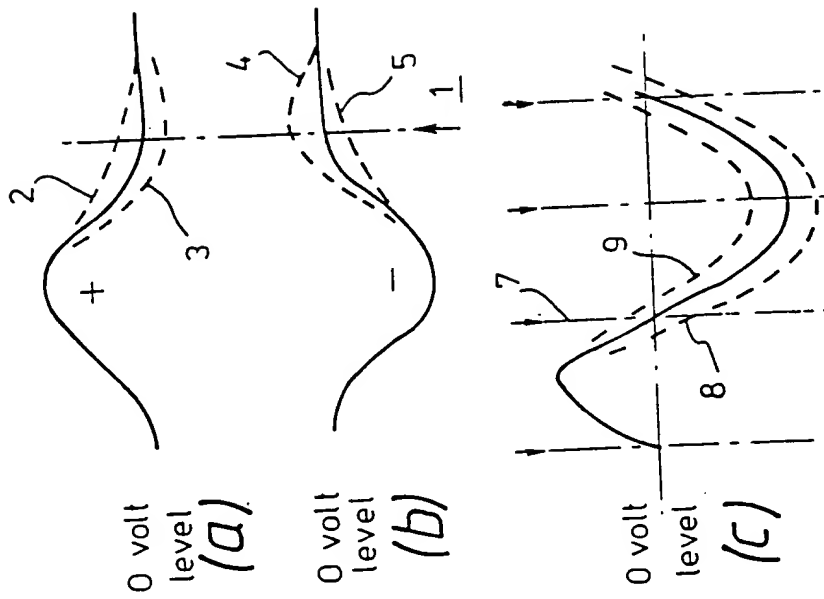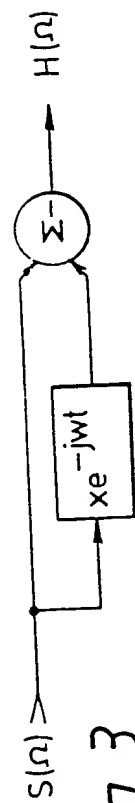
Fig. 4.

0 volt level (a)

+

2

3

0 volt level (b)

4

5

1

-

0 volt level (c)

7

9

8

Fig.2.

(a)

(b)

(c)

Fig.1.

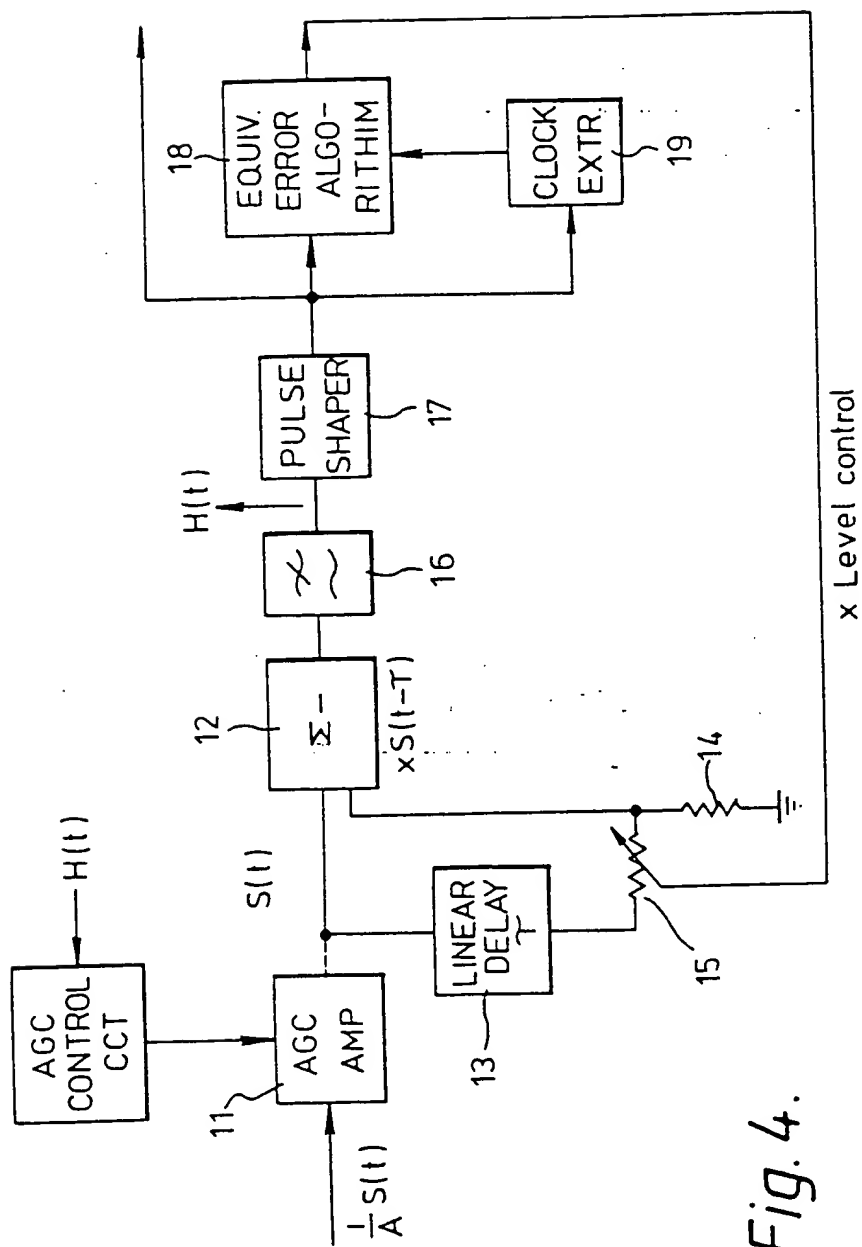$s(\Omega)$ →

$xe^{-j\omega t}$

Σ

→ $H(\Omega)$
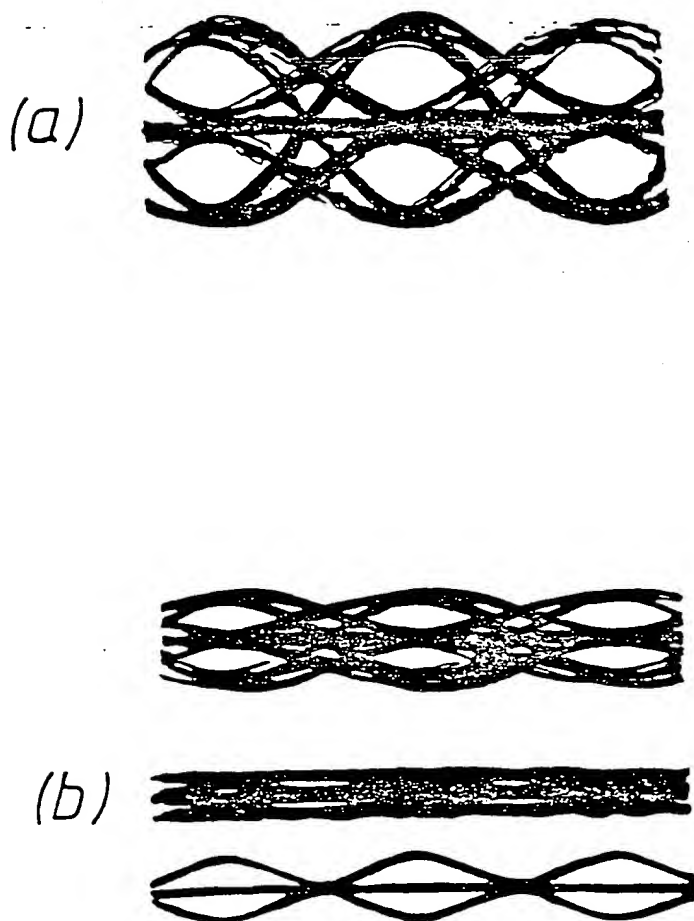
Fig.3.

Fig. 4.

(a)

(b)

Fig. 5.

SPECIFICATION

Adaptive equaliser

5   This invention relates to an adaptive equaliser for
use in a digital line transmission system.
    Pulse transmission down telephone twisted pair
cable is needed when speech is PCM encoded. For
A-law PCM the basic information rate is 64K bits/sec;
10 in digital local area telephony the need for additional
bits for synchronisation and signalling increases this
to 80K bits/sec. For duplex operation on one pair of
wires, a two-four wire hybrid technique can be used,
in which case the line transmission rate remains at
15 80K bits/sec, or time separation can be used. In this
case a number of PCM words, e.g. a whole frame, is
assembled and sent out at a higher rate, usually of
the order of three times the normal bit rate, with the
wires conveying the bit bursts alternately in different
20 direction.
    Since the relatively high bit rates have to be
transmitted over twisted pair cables optimised for
audio band transmission, the pulses would be con-
siderably attenuated and distorted at the end of the
25 line. The waveform shown in Fig. 1a is an example
showing the distortion to a 2 μsec pulse after recep-
tion over 4Km of 0.5 mm transmission cable. The
waveform is amplified and shows that the leading
edge is fairly "well behaved", but the trailing edge
30 has a very slow decay, which can cause inter-symbol
interference. For a continuous mode of transmis-
sion, where pulses of either polarity, or zero, may be
transmitted consecutively, it is desirable to constrain
the period of the pulses to within two periods, i.e. to
35 be zero at the centre of the adjacent pulses and at a
maximum at its centre.
    An object of the invention is to provide an equal-
iser in which the above desideratum is achieved.
    According to the present invention there is pro-
40 vided an adaptive delay equaliser for use in the
reception of intelligence conveyed in pulse form, in
which on reception the pulses are each converted
and delayed by a preset time, and in which the
delayed and inverted pulse is reduced in amplitude
45 by a variable amount and substracted from the orig-
inal pulse, the amount of the time delay and the
amount of the amplitude reduction being such that
the distortion present at the trailing edge of the pulse
is substantially reduced or eliminated.
50   An embodiment of the invention will now be
described with reference to the accompanying draw-
ings, in which
    Figs 1 and 2 are waveform diagrams used to
explain the invention;
55   Fig 3 is a simple diagram explanatory of the prin-
ciple of delay feedback equalisation.
    Fig 4 is a simplified block diagram of an adaptive
delay equaliser embodying the invention.
    The basis of the equaliser will be clear from the
60 waveform of Fig. 1. As already mentioned, Fig. 1a
shows one example of a pulse as received with dis-
tortions due to line conditions. Note especially that
the most serious distortion is the slow delay of the
pulse trailing edge. To minimise or eliminate this
65 distortion, a fraction of the main signal S (t) is

delayed by a period τ, i.e. to give xS(t–τ), where $0 < x
<1$, and subtracted from the main signal. This frac-
tion is shown inverted at Fig. 1b, and the result of the
subtraction is shown at Fig. 1c. The period τ and the
70 fraction x are so chosen that the centre of the main
pulse remains undisturbed and the point which cor-
responds to the centre of an adjacent pulse reaches
zero.
    Although the pulse duration is given as 2 μsec it is
75 assumed that the pulse repetition rate is 4 μsec.
Thus the optimum duration for τ would lie between a
half and one bit period, 2 to 4 τsec in this case,
dependent on the rising edge of the received pulse.
Normally to optimise bandwidth we have found it
80 desirable for the delay to be one bit. The fraction of
the delay signal required depends directly on the
degree of distortion, and equals that portion of the
main signal which remains 4 μsec from the peak. The
technique of pulse equalisation depends on the
85 degree of distortion and is independent of absolute
amplitude.
    The system whose principle is described above is
made adaptive to compensate for varying degrees of
distortion, i.e. line type and length, by devising an
90 algorithm which brings to zero the residue of any
given pulse at the centre of the next pulse. This is
illustrated for ternary transmission, i.e. Alternate
Mark Inversion (AMI) code:

<table>
<tr><td>95</td><td>Previous Pulse
Polarity</td><td>Present Pulse
Polarity</td><td>Correction</td></tr>
<tr><td></td><td>0</td><td>0</td><td>–</td></tr>
<tr><td></td><td>+</td><td>–</td><td>–</td></tr>
<tr><td></td><td>+</td><td>0</td><td>✓</td></tr>
<tr><td>100</td><td>–</td><td>+</td><td>–</td></tr>
<tr><td></td><td>–</td><td>0</td><td>✓</td></tr>
<tr><td></td><td>0</td><td>+</td><td>–</td></tr>
<tr><td></td><td>0</td><td>–</td><td>–</td></tr>
</table>

105   Thus it will be seen that correction is only made
from positive or negative pulses to zero. A simple
data scrambling technique ensures that sufficient
error correcting periods exist which can be averaged
over a suitable time.
110   This correction algorithm is illustrated schemati-
cally in Fig. 2, in which Fig. 2a is for a positive pulse
followed by zero and Fig. 2b is for a negative pulse
followed by a zero. Hence the line 1 corresponds to
the centre of the next bit period and the full line
115 curve is a correctly equalised pulse. The line 2 shows
the state in which the pulse is under-equalised, so
that x has to be increased. The line 3 represents the
case in which the pulse is over-equalised so x has to
be decreased. Similarly in Fig. 2b, the line 4 shows
120 the pulse over-equalised so x must be decreased
while the line 5 shows the pulse under-equalised so
x must be increased. Thus to summarise, for + to –
or – to + one increases x, and for + to – or – to +
one decreases x.
125   For two-level codes such as the Miller code or the
dipulse code, the correction algorithm is simpler
since if there is a transition at the bit boundary 7, Fig.
2c, it will be either early as at 8 if over-equalised or
late as at 9 if under-equalised.
130   An equaliser embodying the above principles was

designed for use with a scrambled AMI code at a bit
rate of 256K bits/sec. and gave a good performance
over a line length of 8 km with 0.5 mm copper dis-
tribution cable – see also the eye diagrams, Fig. 5.
5   The principle of delay feedback equalisation is
illustrated schematically in Fig. 3, as defined by the
following equation:

$$H(\ ) = S(\ )\{1 - xe^{-i\ \cdot}\}., \text{ where } 0 < x < 1$$

10
We now turn to the simplified block diagram of
Fig. 4 to give a more detailed description of an equal-
iser based on the above principles.
In the block diagram of Fig. 4, the incoming signal
15 is defined as $\frac{1}{A}$ S(t), the $\frac{1}{A}$ indicating that it has been
attenuated during its traversal of the line. This signal
is applied to an AGC amplifier 11, whose output is
the amplified signal S(t); the AGC feature ensures
20 that the output of this amplifier has a substantially
constant amplitude. The signal S(t) is applied
directly to one input of a subtraction circuit 12, and
indirectly thereto via a linear delay circuit 13 which
introduces the delay τ. The connection from the
25 delay circuit 13 to the other input of the subtraction
circuit 12 is via an attenuator represented by a fixed
resistor 14 and an adjustable resistor 15.
The output of the subtraction circuit passes via a low
pass filter 16 whose output forms the output of the
30 equaliser, to a pulse shaper 17. This output is applied
to an equalisation error algorithm circuit 18 and to
the clock extraction circuit 19. The latter circuit con-
trols the circuit 18, whose output is indicative of the
error condition of the received signal, and is used to
35 adjust the attenuator, as shown by the connection to
the variable resistor 15. Thus the value x is varied in
the manner referred to above.
By the technique described above it would also be
possible to cancel out forward delay echoes due to
40 bridged tap discontinuities along the transmission
line. This would need at least one more tap delayed
by 2τ and a separate x control. The error algorithm
would then need correction for two consecutive
zeroes, making use of an interative approach.
45 We now refer briefly to Figs. 5a and b which are
derived from photographs of eye diagrams for the
delay feedback equaliser operating our 8Km if
0.5mm distribution cable. Data is sent in the basit
mode at 252K bits/sa., using scrambled AMI code,
50 and the time base gives 1 μs/diy. Fig. 5a shows the
eye at the receiver after automatic gains control and
equalisation. Fig. 5b in the upper row shows the
received eye after equalisation, in the middle row the
received signal before equalisation, and in the bot-
55 tom row the transmit signal.
CLAIMS
1.   An adaptive delay equaliser for use in the
reception of intelligence conveyed in pulse form, in
which on reception the pulses are each inverted and
60 delayed by a preset time, and in which the delayed
and inverted pulse is reduced in amplitude by a vari-
able amount and subtracted from the original pulse,
the amount of the time delay and the amount of the

tially reduced or eliminated.
2.   An equaliser as claimed in claim 1, in which
the intelligence is conveyed in a ternary form so that
the possible line conditions are a positive pulse, a
70 zero state and a negative pulse, and in which the said
delay and subtraction are only effected when a posi-
tive pulse is followed by a zero state or a negative
pulse is followed by a zero state.
3.   An adaptive equaliser for use in the reception
75 of intelligence conveyed in pulse form, substantially
as described with reference to the accompanying
drawings.

New Claims or amendments to claims filed on 1 Dec.
80 1981.
Superseded claims all.
New claims:– 1 to 4

1.   An adaptive delay equaliser for use in the
85 reception of intelligence conveyed in pulse form, in
which in reception the pulses are each delayed by a
preset time, and in which the delayed pulse is
reduced in amplitude by a variable amount and sub-
tracted from the original pulse, the amount of the
90 time delay and the amount of the amplitude reduc-
tion being such that the distortion present at the trail-
ing edge of the pulse is substantially reduced or
eliminated.
2.   An equaliser as claimed in claim 1, in which
95 the intelligence is conveyed in a ternary form so that
the possible line conditions are a positive pulse, a
zero state and a negative pulse, and in which the said
delay and subtraction are only effected when a posi-
tive pulse is followed by a zero state or a negative
100 pulse is followed by a zero state.
3.   An adaptive equaliser for use in the reception
of intelligence conveyed in pulse form, substantially
as described with reference to the accompanying
drawings.
105 4.   An adaptive delay equaliser for use in the
reception of intelligence conveyed in pulse form at a
substantially constant pulse repetition frequency, in
which the pulses as received have been subjected to
distortion producing a slow decay of the trailing
110 edge of a pulse, which slow decay causes interfer-
ence (hereinafter referred to an inter-symbol inter-
ference) with the next received pulse, in which on
reception the pulses are each applied via an amp-
lifier to a first input of a subtraction circuit, in which
115 the pulses as amplified by the amplifier are also
applied via a delay circuit and an attenuator to a
second input of the subtraction circuit, in which the
delay introduced by the delay circuit is such that
pulse as delayed is in synchronism with the next
120 pulse to be received, and in which the attenuator is
so adjusted as to set the amplitude of the pulse
applied to the second input of the subtraction circuit
to a level substantially equal to that of the interfer-
ence due to the slow decay of the pulse's trailing
125 edge, whereby the output from the subtraction cir-
cuit is a pulse sequence from which inter-symbol
interference has been substantially eliminated.